Amendments to the Specification:

Please replace the paragraph beginning on page 10, line 14 with the following rewritten paragraph:

Grid artifact attenuation algorithm proposed in the present invention comprises several procedures. If grid was detected and its orientation and frequency are known then a digital 1-D notch filter is designed as a function of grid frequency and attenuation level. Different algorithms for notch filter design and implementation are known (see: Hamming R. W. "Digital filters", Englewood Cliffs, NJ: Prentice-Hall, 1985, pages 109-118). The goal here is to choose the filter that maximizes the suppression of grid artifacts with minimal image distortion. Both factors mentioned above depend on filter transfer function features such as attenuation level and Gibbs event amplitudes respectively. Attenuation level in its turn depends on filter operator length and bandwidth.

Please replace the paragraph beginning on page 10, line 24 with the following rewritten paragraph:

In the present invention the Potter finite impulse response trigonometric trapezoidal filter algorithm (see: Hamming R. W. "Digital filters", Englewood Cliffs, NJ: Prentice-Hall, 1985, pages 136-140; and Potter R. W. "Compilation of time windows and time shapes for Fourier analysis", 02-5952-0705, Hewlett-Packard, 1971) is proposed as one of the best candidates for the optimal implementation of the notch filter transfer function. The notch filter coefficients are calculated from 2 lowpass filters coefficients tuned on low f_l and high f_2 cut frequencies, which are obtained from grid frequency f_g and bandstop width B using:

Please replace the paragraph beginning on page 11, line 11 with the following rewritten paragraph:

where M is half of the filter symmetrical operator length and ω_n is one of the cut frequencies defined in (1), $\cancel{\omega_n = f_n/(2\pi)}$ $\underline{\omega_n = 2\pi f_n}$ are low cut frequencies, and x is the sampling rate. After that, a smoothing window is applied:

Please replace the paragraph beginning on page 12, line 6 with the following rewritten paragraph:

Attenuation levels can be adjusted using 2 filter parameters: half of the operator length in a range of 24-32 coefficients and rejection bandwidth in a range of 0.07-0.1 as a fraction of Nyquist frequency. Finally, attenuation steps can be designed as a user-selectable option within a range of about -20 dB to –60 dB with increment of -3 to -4 dB. Automatic selection of attenuation level is based on grid peak SNR revealed by the detector. The preferred (default) value can be set up manually for the appropriate attenuation of grid artifacts for a specific CR/DR modality image. Among other possible filter algorithm implementations are Chebysheb filter of type II, and Kaiser filter (see: Hamming R.W. "Digital filters", Englewood Cliffs, NJ: Prentice-Hall, 1985, pages 167-187 and 233-238).

Please replace the paragraph beginning on page 12, line 16 with the following rewritten paragraph:

After digital filter is designed (Fig. 3, box 70), there is a loop for image profiles filtering: horizontal or vertical image profiles are input (box 72), pre-convolution, fast convolution, and post convolution procedures are applied (boxes 74, 76, 78) to each image profile to produce an output image profile (box 82). When all profiles n = N are processed (diamond 80), an output image 83 results. Filtering is based on 1-D spatial domain convolution of filter coefficients, with each of *N* horizontal or vertical (or both for crossed grids) image profile, depending on grid orientation and configuration. *N* is equal to total number of horizontal or vertical profiles in a specific image. The traditional method of convolution is well known in the art of digital image processing (see: Hamming R. W. "Digital filters", Englewood Cliffs, NJ: Prentice-Hall, 1985; and William K. Pratt, "Digital Image Processing", John Wiley & Sons Inc, 1991, pages 171-191).

Please replace the paragraph beginning on page 12, line 27 with the following rewritten paragraph:

In this invention, a technique for the best convolution performance – the main time critical operation - is proposed based on known algorithms such as the "~~fast~~ overlap-save convolution" algorithm (see: ~~Fuller W. A. "Introduction to Statistical Time Series", 2nd Edition, Wiley, John & Sons, Incorporated, 1995~~ Oppenheim et al, "Digital Signal Processing", Prentice-Hall, Inc. 1975, pages, 113-115). This is generally possible with any computer platform, or may also be implemented using low-level vector algebra and math functions libraries for specific computer platforms such as Intel Signal Processing Library (see: Intel SPL, 2000) for Intel based computers. Using the proposed techniques the convolution performance can be improved by 4 to 10 times or more compare to a conventional convolution approach.